

No: 126EK

R13**JAWAHARLAL NEHRU TECHNOLOGICAL UNIVERSITY HYDERABAD****B.Tech III Year II Semester Examinations, May - 2016****DIGITAL SIGNAL PROCESSING****(Common to ECE, EIE)****Time: 3 hours****Max. Marks: 75**

Note: This question paper contains two parts A and B.
 Part A is compulsory which carries 25 marks. Answer all questions in Part A. Part B consists of 5 Units. Answer any one full question from each unit. Each question carries 10 marks and may have a, b, c as sub questions.

PART - A**(25 Marks)**

- 1.a) Write four advantages of Digital Signal Processing over Analog Signal Processing. [2]
- b) Show that the frequency response of a discrete system is a periodic function of frequency. [3]
- c) Give the relation between DTFT and Z-Transform. [2]
- d) Distinguish between Linear convolution and circular convolution. [3]
- e) What are the advantages of Butterworth filter? [2]
- f) What are the advantages and disadvantages of Chebyshev filter. [3]
- g) Define Impulse Response. [2]
- h) Define sampling and Nyquist Rate. [3]
- i) Define Decimation. [2]
- j) What is the need for Multirate Digital Signal Processing? [3]

PART - B**(50 Marks)**

- 2.a) Test the following systems for linearity, time invariance, causality and stability.
 $y(n) = \sin(2\pi n/F)x(n)$
- b) A digital system is characterized by the following difference equation:
 $Y(n) = x(n) + ay(n-1)$ Assuming that the system is relaxed initially, determine its impulse response. [5+5]

OR

3. By taking an example compute DFT by using Over-Lap save method. [10]
- 4.a) Compute the circular convolution of the sequences
 $x_1(n) = \{1, 2, 0, 1\}$ and
 $x_2(n) = \{2, 2, 1, 1\}$ Using DFT approach.
- b) What is FFT? Calculate the number of multiplications needed in the calculation of DFT using FFT algorithm with 32 point sequence. [5+5]

OR

- 5.a) Prove the following properties.
 - i) $x^*(n) \rightarrow X^*((-K))_N R_N(K)$
 - ii) $x^*((-n))_N R_N(n) \rightarrow X_{ep}(k) = \frac{1}{2} [X((K))_N + X^*((-K))_N] R_N(K)$
- b) Compare FFT for the sequence.
 $x[n] = \{1, 0, 1, 1, 0, 1, 1, 1\}$ [5+5]

a) Discuss in detail about spectral transformations.

b) Explain how IIR digital filters are designed from analog filters.

[5+5]

OR

a) Compare the impulse invariance and bilinear transformation methods.

b) Find the order and poles of a low pass Butterworth filter that has a -3db bandwidth of 400 Hz and an attenuation of 20db at 1KHz.

[4+6]

8.a) Draw and explain frequency response of FIR digital filter.

b) Design a high pass filter using hamming window with a cut-off frequency of 1.2 radians/second and $N=9$.

[5+5]

OR

9.a) List the designing steps of FIR filters using fourier method.

b) Design a low pass digital FIR filter using Kaiser Window satisfying the specifications given below.

Pass band cut-off frequency = 100 Hz.

Stop band cut-off frequency = 200 Hz.

Pass band ripple = 0.1dB

Stop band attenuation = 20 dB

Sampling frequency = 1000 Hz.

[3+7]

10.a) What are the Dead band Effects? Discuss.

b) What is mean by sampling rate conversion? Explain.

[5+5]

OR

11.a) What are Limit Cycles and discuss various types of Limit Cycles in brief.

b) Discuss the process of performing sampling rate conversion by an rational factor I/D.

[6+4]